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ABSTRACT OF THE DISCLOSURE

A system and method for processing of audio and speech signals is disclosed, which provide compatibility over a range of communication devices operating at different sampling frequencies and/or bit rates. The analyzer of the system divides the input signal in different portions, at least one of which carries information sufficient to provide intelligible reconstruction of the input signal. The analyzer also encodes separate information about other portions of the signal in an embedded manner, so that a smooth transition can be achieved from low bit-rate to high bit-rate applications. Accordingly, communication devices operating at different sampling rates and/or bit-rates can extract corresponding information from the output bit stream of the analyzer. In the present invention embedded information generally relates to separate parameters of the input signal, or to additional resolution in the transmission of original signal parameters. Non-linear techniques for enhancing the overall performance of the system are also disclosed. Also disclosed is a novel method of improving the quantization of signal parameters. In a specific embodiment the input signal is processed in two or more modes dependent on the state of the signal in a frame. When the signal is determined to be in a transition state, the encoder provides phase information about N sinusoids, which the decoder end uses to improve the quality of the output signal at low bit rates.